## IN THE CLAIMS

**Please amend the claims as follows:**

Claim 1 (Currently Amended) A ~~M~~ method for processing speech, comprising:

~~comprising the steps of:~~

[[-]] receiving a speech input ~~(SI)~~ of a speaker ~~(S)~~,

[[-]] generating speech parameters ~~(SP)~~ from said speech input ~~(SI)~~,

[[-]] determining parameters describing an absolute loudness ~~(L)~~ of said speech input ~~(SI)~~, and

[[-]] evaluating ~~(EV)~~ at least one of said speech input ~~(SI)~~ and [[/or]] said speech parameters ~~(SP)~~ using said parameters describing the absolute loudness ~~(L)~~.

Claim 2 (Currently Amended) The ~~M~~ method according to claim 1, wherein the step of evaluation ~~(EV)~~ comprises a step of emotion recognition.

Claim 3 (Currently Amended) The ~~M~~ method according to claim 1, wherein the step of evaluation ~~(EV)~~ comprises a step of speaker identification.

Claim 4 (Currently Amended) The ~~M~~ method according to claim 1, wherein a microphone array ~~(MA)~~ comprising a plurality of microphones is used for determining said parameters describing the absolute loudness ~~(L)~~.

Claim 5 (Currently Amended) The ~~M~~ method according to claim 1, wherein at least one of a location and [[/or]] distance ~~(D)~~ of the speaker is determined.

Claim 6 (Currently Amended) The ~~M~~ method according to claim 1, wherein

~~characterized in that~~

the absolute loudness ~~(L)~~ is determined using algorithms for at least one of auditory and [[/or]] binaural processing.

Claim 7 (Currently Amended) The ~~M~~ method according to claim 5, wherein

~~characterized in that~~

said absolute loudness ~~(L)~~ is computed by normalizing a measured loudness, or energy by said distance ~~(D)~~.

Claim 8 (Currently Amended) The ~~M~~ method according to claim 5, wherein

~~characterized in~~

~~that~~ said distance ~~(D)~~ is determined using the time delay ~~(TD)~~ of the speech input between said plurality of microphones.

Claim 9 (Currently Amended) A ~~S~~ speech processing system,

which is ~~capable of performing or realizing a method for processing speech according to claim 1 and/or the steps thereof~~ configured to:

receive a speech input of a speaker,

generate speech parameters from said speech input,

determine parameters describing an absolute loudness of said speech input, and

evaluate at least one of said speech input and said speech parameters using said parameters describing the absolute loudness.

Claim 10 (Cancelled).

Claim 11 (Cancelled).

Claim 12 (New) A computer readable medium encoded with a computer program configured to cause a processor-based device to execute a method of:

receiving a speech input of a speaker,

generating speech parameters from said speech input,

determining parameters describing an absolute loudness of said speech input,

evaluating at least one of said speech input and said speech parameters using said parameters describing the absolute loudness.

Claim 13 (New) A method for processing speech, comprising:

receiving a speech signal of a speaker;

generating speech parameters from said speech signal;

determining a distance of the speaker based on a time delay of a respective arrival of said speech signal at two or more microphones;

normalizing a measured loudness or energy by said distance; and

evaluating at least one of said speech signal and said speech parameters using the normalized loudness or energy.

Claim 14 (New) A system for emotion recognition and/or speaker identification , comprising:

at least two microphones configured to receive a speech signal;

a data processor configured to generate speech parameters from said speech signal, to determine a distance of the speaker based on a time delay of a respective arrival of said

speech signal at said microphone, to normalize a measured loudness or energy by said distance, and

further configured to evaluate at least one of said speech signal and said speech parameters using the normalized loudness or energy.